AMENDMENT(S) TO THE CLAIMS

1. (canceled).

2. (amended) The method as defined in Claim ~~1~~ 4, further comprising:

receiving the encoded video object plane at the receiver from the connection;

demultiplexing the encoded video object plane into coded video and audio streams;

inputting the coded video and audio streams, respectively, into video and audio decoders;

inputting the decoded video and audio streams to a media mixer; and

inputting the mixed video and audio streams output from the media mixer to an output device.

3. (canceled).

4. (amended) A method for transmitting a mixed media data stream in packets, including audio and video objects, between a sender and a receiver through a connection over a network, the method comprising:

monitoring, at the receiver, transmission characteristics of the connection between the server and the receiver;

estimating available bandwidth at the sender based upon the transmission characteristics of the connection monitored at the receiver;

allocating a global buffer for the mixed media data stream to be transmitted from the sender to the receiver as a function of the estimated available bandwidth at the sender;

pre-encoding a portion of each Video Object Plane (VOP) in the global buffer with respect to a quantization parameter (QP) of the VOP;

encoding the VOP in the global buffer based on the QP;

updating a rate distortion model based upon the QP and packet loss rate;

performing a frame skipping function after the VOP encoding; and

transmitting from the sender to the receiver the encoded video object plane in the global buffer at a regulated sender transmission rate from the sender as a function of the estimated available bandwidth at the sender;

wherein pre-encoding a portion of each VOP with respect to the QP of the VOP further comprises adjusting the QP of the VOP; and

~~The method as defined in Claim 3,~~ wherein the QP of the VOP is adjusted with respect to a texture parameter, r, as the number of bits which will be used to encode the VOP wherein:

$$r = \frac{p_1 \times MAD}{QP} + \frac{p_2 \times MAD}{QP^2};$$

p_1 and p_2 are control parameters; and

MAD is a mean absolute distortion in which a total target bit rate for all objects in the global buffer are allocated proportionally to motion, size, and square of MAD.

5. (original) The method as defined in Claim 4, wherein the adjusting of the QP of the VOP is performed by changing the QP to a values in a range from 1 to 31 depending upon the estimated available bandwidth at the receiver.

6. (amended) A method for transmitting a mixed media data stream in packets, including audio and video objects, between a sender and a receiver through a connection over a network, the method comprising:

monitoring, at the receiver, transmission characteristics of the connection between the server and the receiver;

estimating available bandwidth at the sender based upon the transmission characteristics of the connection monitored at the receiver;

allocating a global buffer for the mixed media data stream to be transmitted from the sender to the receiver as a function of the estimated available bandwidth at the sender;

pre-encoding a portion of each Video Object Plane (VOP) in the global buffer with respect to a quantization parameter (QP) of the VOP;

encoding the VOP in the global buffer based on the QP;

updating a rate distortion model based upon the QP and packet loss rate;

performing a frame skipping function after the VOP encoding; and

transmitting from the sender to the receiver the encoded video object plane in the global buffer at a regulated sender transmission rate from the sender as a function of the estimated available bandwidth at the sender; and

~~The method as defined in Claim 1,~~ wherein updating a rate distortion model based upon the QP and packet loss rate comprises:

predicting the number of bits, r_i, to encode the ith VOP, and is given by:

$$r_i = \frac{(p_1)_i \times MAD_i}{QP_i} + \frac{(p_2)_i \times MAD_i}{QP_i^2};$$

the distortion, d, is estimated by

$$d_i = (q_1)_i \times QP_i + (q_2)_i \times QP_i^2 + (q_3)_i \times r_i \times (P_L)_i, \quad \text{wherein:}$$

q_1, q_2 and q_3 are control parameters; and

the packet loss rate $(P_L)_i$ is an estimate of that the probability that the ith transmission of data from the sender will be lost; and

minimizing the overall distortion, D, for each encoded VOP by $D = \sum_i d_i$, subject to $R = \sum_i r_i \leq R_T$, where R_T is the total bit budget for the current time instant.

7. (canceled).

8. (amended) A method for transmitting a mixed media data stream in packets, including audio and video objects, between a sender and a receiver through a connection over a network, the method comprising:

monitoring, at the receiver, transmission characteristics of the connection between the server and the receiver;

estimating available bandwidth at the sender based upon the transmission characteristics of the connection monitored at the receiver;

allocating a global buffer for the mixed media data stream to be transmitted from the sender to the receiver as a function of the estimated available bandwidth at the sender;

pre-encoding a portion of each Video Object Plane (VOP) in the global buffer with respect to a quantization parameter (QP) of the VOP;

encoding the VOP in the global buffer based on the QP;

updating a rate distortion model based upon the QP and packet loss rate;

performing a frame skipping function after the VOP encoding; and

transmitting from the sender to the receiver the encoded video object plane in the global buffer at a regulated sender transmission rate from the sender as a function of the estimated available bandwidth at the sender; and

wherein:

the sender sends data to the receiver in through a connection over a packet switched network in a sender packet having a sender header that includes:

a packet sequence number;

a timestamp indicating the time when the sender packet was sent (ST1); and

the size of the sender packet (PacketSize);

the receiver sends data to the sender through the connection over the packet switched network in a receiver packet having a receiver header that includes:

the time interval that the sender packet spent in the receiver side (ΔRT);

the timestamp of the sender packet sent from the sender (ST1);

an estimate, calculated by the receiver, of a packet-loss rate; and

the rate at which data is received at the receiver;

monitoring transmission characteristics of the connection between server and receiver comprises:

estimating a round trip time of the sender packet from the sender to the receiver (RTT) based on ST1 and ΔRT;

estimating a time out interval (TO) before which the sender should retransmit to the receiver a sender packet of data that has not been received by the receiver;

estimating a probability that a packet of data will be lost (P_L);

estimating the present available network bandwidth at which the receiver can receive data from the sender (rvcrate) as a function of the PacketSize, the RTT, the P_L, and the TO;

deriving the present sending rate of data from the sender to the receiver ($\overline{currate}$);

setting an updated sending rate of data from the sender to the receiver (currate), wherein:

if rcvrate is greater than $\overline{currate}$, then deriving currate as a function $\overline{currate}$, PacketSize, and RTT; and

<u>if rcvrate is not greater than $\overline{currate}$, then setting currate to be less than rcvrate;</u>

<u>and</u>

~~The method as defined in Claim 7,~~ wherein:

$RTT = \alpha \times \overline{RTT} + (1-\alpha) \times (now - ST1 - \Delta RT)$; and

now is the timestamp indicating the time at which the receiver packet was received in the sender; and α is a weighting parameter.

9. (amended) A method for transmitting a mixed media data stream in packets, including audio and video objects, between a sender and a receiver through a connection over a network, the method comprising:

monitoring, at the receiver, transmission characteristics of the connection between the server and the receiver;

estimating available bandwidth at the sender based upon the transmission characteristics of the connection monitored at the receiver;

allocating a global buffer for the mixed media data stream to be transmitted from the sender to the receiver as a function of the estimated available bandwidth at the sender;

pre-encoding a portion of each Video Object Plane (VOP) in the global buffer with respect to a quantization parameter (QP) of the VOP;

encoding the VOP in the global buffer based on the QP;

updating a rate distortion model based upon the QP and packet loss rate;

performing a frame skipping function after the VOP encoding; and

transmitting from the sender to the receiver the encoded video object plane in the global buffer at a regulated sender transmission rate from the sender as a function of the estimated available bandwidth at the sender; and

wherein:

the sender sends data to the receiver in through a connection over a packet switched network in a sender packet having a sender header that includes:

a packet sequence number;

a timestamp indicating the time when the sender packet was sent (ST1); and

the size of the sender packet (PacketSize);

the receiver sends data to the sender through the connection over the packet switched network in a receiver packet having a receiver header that includes:

the time interval that the sender packet spent in the receiver side (ΔRT);

the timestamp of the sender packet sent from the sender (ST1);

an estimate, calculated by the receiver, of a packet-loss rate; and

the rate at which data is received at the receiver;

monitoring transmission characteristics of the connection between server and receiver comprises:

estimating a round trip time of the sender packet from the sender to the receiver (RTT) based on ST1 and ΔRT;

estimating a time out interval (TO) before which the sender should retransmit to the receiver a sender packet of data that has not been received by the receiver;

estimating a probability that a packet of data will be lost (P_L);

estimating the present available network bandwidth at which the receiver can receive data from the sender (rvcrate) as a function of the PacketSize, the RTT, the P_L, and the TO;

deriving the present sending rate of data from the sender to the receiver ($\overline{currate}$);

setting an updated sending rate of data from the sender to the receiver (currate), wherein:

if rcvrate is greater than $\overline{currate}$, then deriving currate as a function $\overline{currate}$, PacketSize, and RTT; and

if rcvrate is not greater than $\overline{currate}$, then setting currate to be less than rcvrate; and

~~The method as defined in Claim 7,~~ wherein:

$TO = RTT + (k \times RTTVAR)$;

k is a constant;

$RTTVAR = \alpha_2 \times \overline{RTTVAR} + (1-\alpha_2) \times \left| RTT - (now - ST1 - \Delta RT) \right|$;

$\overline{RTTVAR}$ is the current variation in the round trip time of the sender packet from the sender to the receiver (RTT);

α2 is a weighting parameter; and

RTTVAR is a smoothed estimate of $\overline{RTTVAR}$.

10. (canceled).

11. (amended) A method for transmitting a mixed media data stream in packets, including audio and video objects, between a sender and a receiver through a connection over a network, the method comprising:

monitoring, at the receiver, transmission characteristics of the connection between the server and the receiver;

estimating available bandwidth at the sender based upon the transmission characteristics of the connection monitored at the receiver;

allocating a global buffer for the mixed media data stream to be transmitted from the sender to the receiver as a function of the estimated available bandwidth at the sender;

pre-encoding a portion of each Video Object Plane (VOP) in the global buffer with respect to a quantization parameter (QP) of the VOP;

encoding the VOP in the global buffer based on the QP;

updating a rate distortion model based upon the QP and packet loss rate;

performing a frame skipping function after the VOP encoding; and

transmitting from the sender to the receiver the encoded video object plane in the global buffer at a regulated sender transmission rate from the sender as a function of the estimated available bandwidth at the sender

wherein:

the sender sends data to the receiver in through a connection over a packet switched network in a sender packet having a sender header that includes:

a packet sequence number;

a timestamp indicating the time when the sender packet was sent (ST1); and

the size of the sender packet (PacketSize);

the receiver sends data to the sender through the connection over the packet switched network in a receiver packet having a receiver header that includes:

the time interval that the sender packet spent in the receiver side (ΔRT);

the timestamp of the sender packet sent from the sender (ST1);

an estimate, calculated by the receiver, of a packet-loss rate; and

the rate at which data is received at the receiver;

monitoring transmission characteristics of the connection between server and receiver comprises:

estimating a round trip time of the sender packet from the sender to the receiver (RTT) based on ST1 and ΔRT;

estimating a time out interval (TO) before which the sender should retransmit to the receiver a sender packet of data that has not been received by the receiver;

estimating a probability that a packet of data will be lost (P_L);

estimating the present available network bandwidth at which the receiver can receive data from the sender (rvcrate) as a function of the PacketSize, the RTT, the P_L, and the TO;

deriving the present sending rate of data from the sender to the receiver ($\overline{currate}$);

setting an updated sending rate of data from the sender to the receiver (currate), wherein:

if rcvrate is greater than $\overline{currate}$, then deriving currate as a function $\overline{currate}$, PacketSize, and RTT; and

if rcvrate is not greater than $\overline{currate}$, then setting currate to be less than rcvrate;

wherein P_L is derived by a Gilbert Model; and

~~The method as defined in Claim 10,~~

wherein:

$$P_L = \frac{\hat{q}}{\hat{p} + \hat{q}};$$

$$\{X_i\}_{i=1}^{n};$$

X_i takes 1 if the ith sender packet has arrived successfully at the receiver;

X_i takes 0 if the ith sender packet is lost;

$$p = P[X_i = 1 \mid X_{i-1} = 0];$$

$$q = P[X_i = 0 \mid X_{i-1} = 1];$$

$\hat{p}$ is an estimate of p;

$\hat{q}$ is an estimate of q; and

$\hat{p} = n_{01}/n_0$ and $\hat{q} = n_{10}/n_1$, wherein:

n_{01} is the number of times in an observed time series when one follows zero;

n_{10} is the number of times when zero follows one;

n_0 is the number of zeros; and

n_1 is the number of ones.

12. (original) The method as defined in Claim 11, wherein:

the P_L is further smoothed by a filter that weights the n most recent measured packet loss rates by:

$$P_{L,i} = \sum_{j=0}^{n-1}(w_j \times \overline{P_{L,i-j}})$$;

$\overline{P_{L,i-j}}$ is the measured packet loss rate in the (i-j)th time interval;

two set of weighting parameters are defined as follows:

	W0	W1	W2	W3	W4	W5	W6	W7
WS1	1.0	1.0	1.0	1.0	0.8	0.6	0.4	0.2

	W0	W1	W2	W3	W4	W5	W6	W7
WS2	1.2	1.2	1.0	1.0	0.8	0.5	0.3	0.1

; and WS2 is used for w_j when the actual packet loss rate is less than half of the measured packet loss rate, otherwise WS1 is used for w_j.

13. (amended) A computer-readable media comprising computer-executable instructions for performing the method as recited in Claim ~~1~~11.

14. (canceled).

15. (amended) A method for transmitting a mixed media data stream in packets, including audio and multiple video objects (MVOs), between a sender and a receiver through a connection over a network, the method comprising:

monitoring transmission characteristics of one or more encoded video object planes through the connection between the sender and the receiver;

estimating, from the transmission characteristics, an available bandwidth (RT) at the sender;

allocating, as a function of the RT, a portion of the mixed media data stream to a global buffer;

encoding a video object plane from the global buffer based upon a rate distortion function that accounts for packet loss rate between sender and receiver;

updating the rate distortion function based upon results of the encoded video object plane and upon a memory containing results of one or more previously encoded video object planes;

after the encoding the MVOs in the video object plane, performing a frame skipping function; and

transmitting, at the estimated available bandwidth, the encoded video object plane from the sender to the receiver;

~~The method as defined in Claim 14,~~

wherein allocating a portion of the mixed media data stream to a global buffer comprises:

$W_{cur} = \max(((W_{prev} + B_{prev}) \times R_T / R_{old} - R_T / F),\ 0)$, as the global buffer size $R_{old}/2$, is changed to $R_T/2$, wherein:

B_{prev} is the number of bits spent in the previous time instant B_{prev},

$R_{old}/2$ is the previous size of the global buffer;

W_{prev} is the previous occupancy of the global buffer; and

F is the video frame rate.

16. (amended) The method as defined in Claim ~~14~~15, wherein allocating a portion of the mixed media data stream to a global buffer comprises the allocation of an output target rate from the global buffer among each of video and audio data streams so as to yield the target bits for an individual object in the data stream.

17. (amended) The method as defined in Claim ~~14~~15, further comprising:

receiving the encoded video object plane at the receiver from the connection;

demultiplexing the encoded video object plane into coded video and audio streams;

inputting the coded video and audio streams, respectively, into video and audio decoders;~~ and~~

inputting the decoded video and audio streams to a media mixer; and

inputting the mixed video and audio streams output from the media mixer to an output device.

18. (amended) A computer-readable media comprising computer-executable instructions for performing the method as recited in Claim ~~14~~15.

19. (canceled).

20. (canceled).

21. (canceled).

22. (canceled).

23. (canceled).

24. (canceled).

25. (canceled).

26. (canceled).

27. (canceled).

28. (canceled).

29. (canceled).

30. (canceled).

31. (canceled).

32. (canceled).

33. (canceled).

34. (canceled).

35. (new) A computer-readable media comprising computer-executable instructions for performing the method as recited in Claim 4.

36. (new) A computer-readable media comprising computer-executable instructions for performing the method as recited in Claim 6.

37. (new) A computer-readable media comprising computer-executable instructions for performing the method as recited in Claim 8.

38. (new) A computer-readable media comprising computer-executable instructions for performing the method as recited in Claim 9.